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(54) Abstract Title
Speech coding in a telecommunication system

(57) An arrangement for setting up a call between first and second mobile telecommunication networks in which the speech codec to be used is negotiated between the two networks. At least one of the networks comprises a tandem free operation (TFO) device outside the radio access network and the negotiation occurs between the TFO devices of the first and second networks. The radio access network is notified of the determined speech codec by sending a call control message from the associated TFO device. The two mobile telecommunications networks may be interconnected by the PSTN. The arrangement helps prevent a degradation in speech quality by allowing the coded speech to be passed between the two networks without further coding and decoding operations.

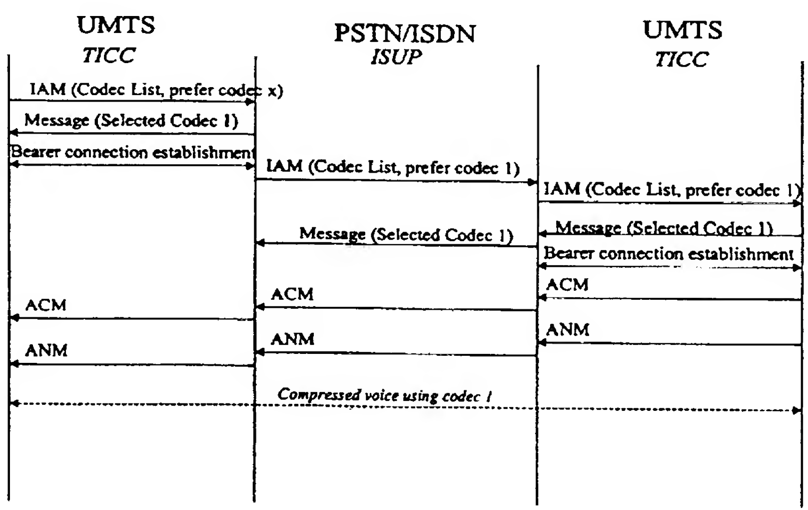

Figure 3

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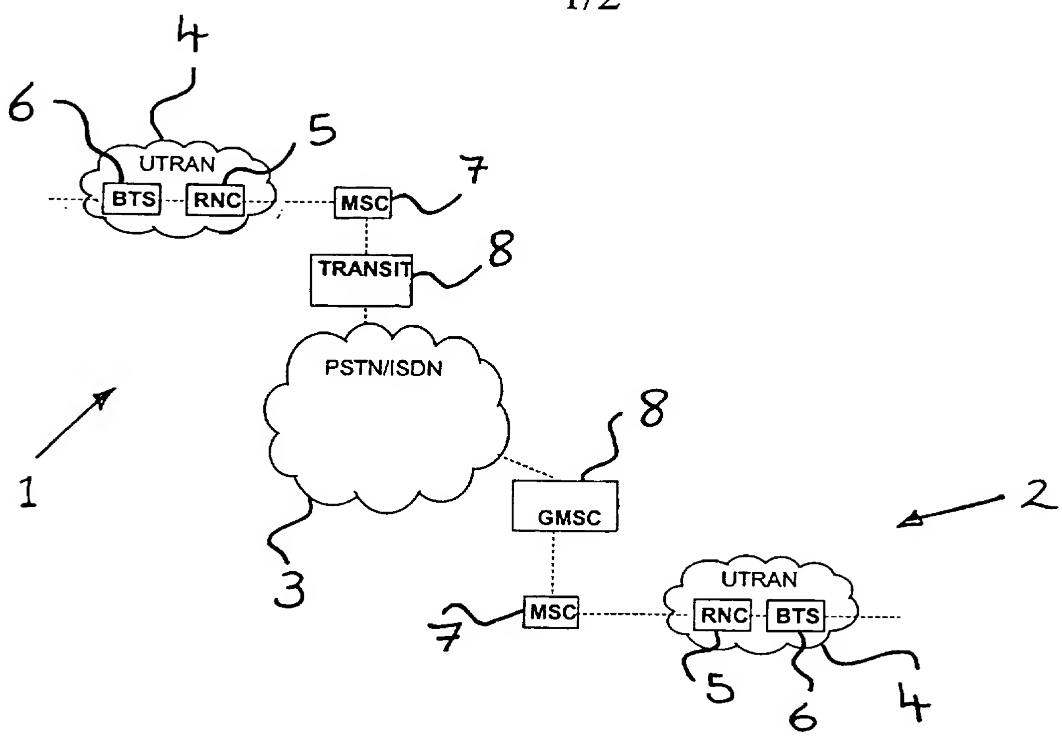

Figure 1

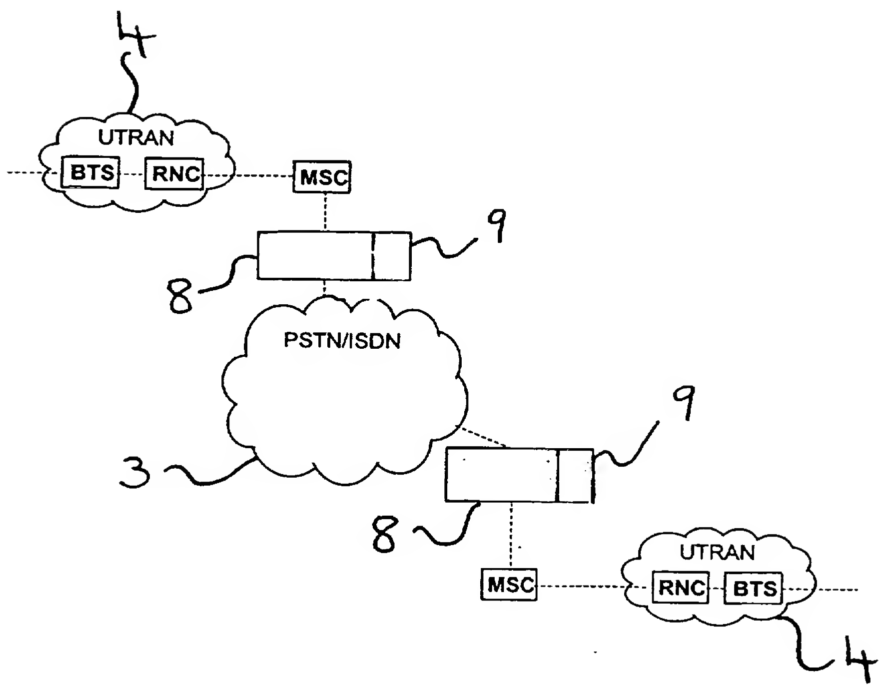

Figure 2

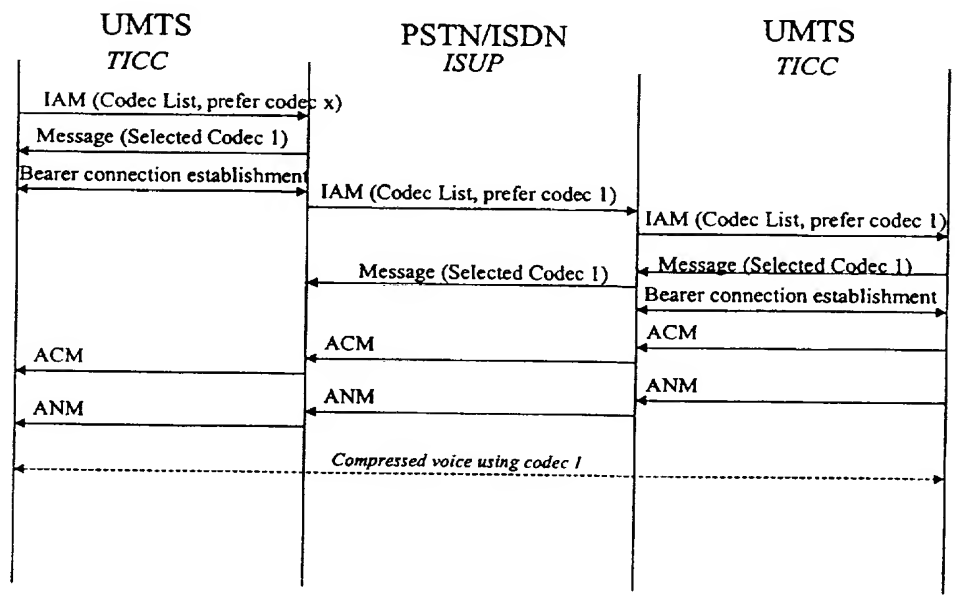

Figure 3

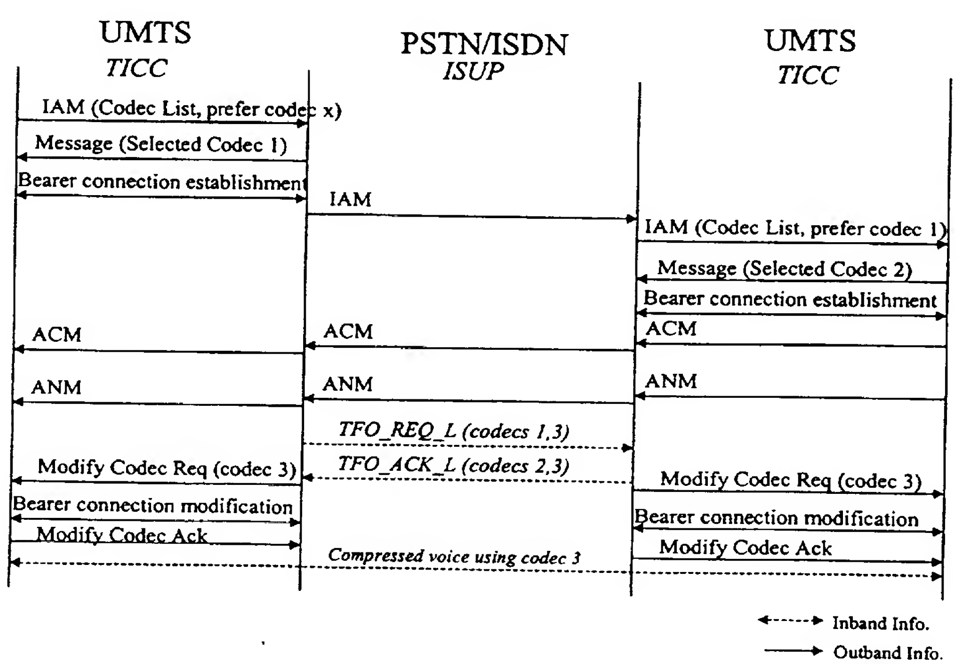

Figure 4

SPEECH CODING IN A TELECOMMUNICATIONS SYSTEM

Field of the Invention

The present invention relates to speech coding in a telecommunications system and more particularly, though not necessarily, to speech coding in a system where speech data is transferred between two mobile telephone networks via a PSTN/ISDN telephone network.

Background to the invention

Conventional Public Switched Telephone Networks (PSTN) digitally encode speech data for transmission using Pulse Code Modulation (PCM). On the other hand, digital mobile telephone networks make use of more advanced coding techniques such as CELP and Adaptive Multi-Rate (AMR) coding, which achieve higher compression ratios than can be achieved with PCM. In many mobile networks, coding and decoding of speech is carried out at the mobile terminals themselves. Providing that a call is made between two mobile terminals both registered with the same network it may be possible to transmit encoded speech data from end to end.

In the event that a call is from a mobile terminal registered with a mobile network, to a terminal which is a subscriber of a "foreign" network, end to end transmission of encoded speech data may not be possible, depending upon the nature of the foreign network and of any intermediate networks which connect the originating mobile network to the foreign network (the same will of course be true where the call originates at the foreign network).

Consider the telecommunications system of Figure 1 which illustrates two third generation Universal Mobile Telecommunication System (UMTS) networks 1,2 which are coupled via a conventional PSTN/ISDN network 3. The UMTS networks 1,2 each comprise a UMTS Terrestrial Radio Access Network (UTRAN) 4 having Radio

Network Controllers (RNCs) 5 and Base Transceiver Stations (BTSs) 6. A UTRAN 5 passes compressed speech data between a mobile terminal (not shown) and a Mobile Switching Centre (MSC) 7 which routes incoming and outgoing call connections.

Assume that a call originates from a subscriber of one of the UMTS networks 1 and is made to a subscriber of the other of the UMTS networks 2. The call is routed via the PSTN 3 using respective Transit nodes 8 - one of which is a Gateway MSCs (GMSCs) - of the UMTS networks 1,2. As has already been noted, the PSTN 3 uses PCM to encode speech data. Now it is important that any speech data transferred through the PSTN 3 is in a form which can be understood by that network. This is necessary, for example, to allow the PSTN 3 to insert operator announcements into a speech call, to perform voice prompting services, etc, as well as to allow the operator of the PSTN 3 to monitor call, e.g. for security purposes. It is therefore necessary to "transcode" speech data at the GMSCs 8 of the UMTS networks 1,2 prior to passing the data to the PSTN 3, i.e. the speech data is converted from a mobile network speech coding format to PCM. Similarly, PCM data received at the GMSCs 8 must be converted to the appropriate mobile network speech coding format

Transcoding consumes considerable processing resources at a GMSC 8 and also results in a perceivable degradation in speech quality. In order to at least partially overcome these disadvantages, Tandem Free Operation (TFO) devices may be introduced into the speech connection at the GMSCs 8. Outgoing speech data continues to be converted to PCM, but the least significant bits of each PCM sample are "stolen" by the TFO device. The stolen bits form a channel which has sufficient bandwidth (i.e. 8Kbits/sec) to carry the original coded data. The TFO device at the terminating UMTS network reassembles the coded data for forwarding to the associated UTRAN whilst the received PCM data is discarded (unless it has been modified by the PSTN, e.g. by the addition of an operator announcement). In this way, TFO makes PCM data available to the PSTN 3, whilst still allowing the end to end transmission of efficiently coded speech data.

In the event that intermediate devices within the PSTN/ISDN alter the PCM bit stream, the TFO devices detect the change and "fall-back" to passing the PCM coded speech between the TFO devices, i.e. they no longer pass on the compressed voice data.

The speech codecs available to a mobile network depend upon the nature of the network and possibly upon the nature of a terminal using the network. It will be obvious that the end to end use of a single codec is only possible when two networks are both capable of using the same codec. Assuming that the GMSCs of two mobile communication networks are aware of the codec capabilities of the networks to which they belong, it is possible for them to negotiate and agree upon a common codec. Indeed, a suitable protocol is provided for in the ETSI recommendation GSM 08.62 (version 7.0.0, release 1998).

A problem arises in attempting to implement TFO in mobile networks such as are illustrated in Figure 1, where the TFO devices are located on the fringe of the mobile networks, i.e. outside of the UTRAN (conventionally TFO devices are located within the radio access networks). There is currently no mechanism for exchanging information, concerning codecs negotiated between TFO devices, between TFO devices and radio access networks, in such networks.

Summary of the Present Invention.

Telecommunications networks currently rely to a large extent upon the Signalling System no.7 (SS7) as the mechanism for controlling call connections and for handling the transfer of signalling information between signalling points of the networks. Typically, one or more Application and User Parts at a given signalling point will make use of SS7 to communicate with peer application and user parts at some other signalling point. Examples of User Parts are ISUP (ISDN User Part) and TUP (Telephony User Part) whilst examples of Application Parts are INAP (Intelligent Network Application Part) and MAP (Mobile Application Part). The conventional SS7 protocol stack includes Message Transfer Parts MTP1, MTP2, and MTP3 which handle the formatting of signalling messages for transport over the physical layer as well as various routing

functions. The conventional physical transport network over which signalling messages are sent is a Synchronous Transfer Mode (STM) network such as E.1 (Europe) or T.1 (America). User plane data, e.g. voice information, is sent over the same STM network.

There has been considerable interest of late amongst the telecommunications community in using non-standard (i.e. non-conventional within the telecommunications industry) bearer transport mechanisms (non-STMs) in telecommunications networks for carrying user plane data. The reasons for this are related both to improvements in efficiency as well as potential cost savings. Much consideration has been given for example to the use of Asynchronous Transfer Mode (ATM) networks to transport signalling information between signalling points.

Typically, the bearer transport mechanism protocol layers lie beneath SS7, and ISUP, which deals with the setting-up and control of call connections, is closely linked to the bearer transport mechanism. ISUP therefore does not readily lend itself to use with non-STM bearer transport technologies. As such, several standardisation bodies including the ITU-T, ETSI, and ANSI, are currently considering the specification of a protocol for the control of calls, which is independent of the underlying bearer transport mechanism. This can be viewed as separating out from the protocol, bearer control functions which relate merely to establishing the parameters (including the start and end points) of the "pipe" via which user plane data is transported between nodes, and which are specific to the transport mechanism. The new protocol, referred to hereinafter as Transport Independent Call Control (TICC), retains call control functions such as the services invoked for a call between given calling and called parties (e.g. call forwarding), and the overall routing of user plane data.

According to a first aspect of the present invention there is provided a method of setting up a call connection between first and second mobile telephone networks at least one of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the method comprising the steps of:

conducting a negotiation between said TFO device of one of the mobile networks and a peer TFO device of the other mobile network to determine a suitable speech codec; and

notifying said Radio Access Network of the determined speech codec by sending a Call Control (CC) message from the associated TFO device.

It may be the case that each of the mobile networks comprises a Tandem Free Operation (TFO) device located outside of the Radio Access network part(s) of the mobile network. However, this need not be the case, and one of the mobile networks may have a TFO device within the radio Access Network.

The two mobile telephone networks may be coupled to one another via a PSTN. Preferably, said negotiation is carried out using the Codec Mismatch Resolution and Optimisation Procedure in the TFO protocol, with TFO messages being sent using in-band signalling.

Preferably, said the or each TFO device located outside of a Radio Access network is located at a Gateway MSC (GMSC) which provides an interface between the mobile network and foreign networks, e.g. a PSTN.

The present invention is particularly suited to setting up a call connection between subscribers, one of whom is a subscriber of a Universal Mobile Telecommunication System (UMTS) network. In this case, the Radio Access Network is preferably a UMTS Terrestrial Radio Access Network (UTRAN), with the TFO device of the network being located on the edge of the UMTS core network.

According to a second aspect of the present invention there is provided apparatus for setting up a call connection between first and second mobile telephone networks at least one of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the apparatus comprising:

means for conducting a negotiation between said TFO device of one of the mobile networks and a peer TFO device of the other mobile network to determine a suitable speech codec; and

means for notifying said Radio Access Network of the determined speech codec by sending a Call Control (CC) message from the associated TFO device.

According to a third aspect of the present invention there is provided a method of setting up a call connection between first and second mobile telephone networks each of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the mobile networks being coupled via a Public Switched Telephone Network (PSTN), the method comprising the step of:

conducting a negotiation between two mobile terminals subscribing to the first and second mobile networks respectively to determine a suitable speech codec, wherein the negotiation is conducted using Call Control protocol signalling messages exchanged between the Radio Access Network parts and the respective TFO devices, and ISUP messages sent between the TFO devices.

For example, a list of codecs which are available to an originating mobile network may be sent from the TFO device of that network to the peer TFO device using an ISUP Initial Address Message (IAM). The message may additionally include the codec preferred by the originating mobile network. A subsequent ISUP message sent in the backward direction will indicate the codec type selected by the terminating mobile terminal. In order to enable this negotiation procedure, it may be necessary to modify the ISUP standard.

According to a fourth aspect of the present invention there is provided apparatus for setting up a call connection between first and second mobile telephone networks each of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the mobile networks being coupled via a Public Switched Telephone Network (PSTN), the apparatus comprising:

means for conducting a negotiation between two mobile terminals subscribing to the first and second mobile networks respectively to determine a suitable speech codec, wherein the negotiation is conducted using Call Control protocol signalling messages exchanged between the Radio Access Network parts and the respective TFO devices, and ISUP messages sent between the TFO devices.

Brief Description of the Drawings

Figure 1 illustrates schematically a telecommunications system of known design;
Figure 2 illustrates schematically a modified telecommunications system;
Figure 3 illustrates a first set of signals associated with codec negotiation in the system of Figure 2 and according to a first embodiment of the present invention; and
Figure 4 illustrates a second set of signals associated with codec negotiation in the system of Figure 2 and according to a second embodiment of the present invention.

Detailed Description of Certain Embodiments

There has already been described with reference to Figure 1 a telecommunication system which makes use of transcoding to allow speech traffic to be transferred between two mobile networks via a PSTN. Figure 2 illustrates a modified system architecture in which the GMSCs of the two UMTS networks have incorporated thereinto TFO devices 9.

Figure 3 illustrates signalling associated with call set-up between two nodes in respective UMTS mobile telephone networks. Nodes 1 and 4 represent MSCs, whilst the two middle Nodes 2 and 3 represent GMSCs.

Call set-up signalling within the UMTS networks is conducted at the Call Control (e.g. TICC) level, call set-up being initiated by an Initial Address Message (IAM) sent from an MSC to the associated GMSC. This IAM uses a Generic Capabilities Negotiation (GCN) mechanism to determine a number of parameters for the call connection. In particular, the IAM contains a list of codecs supported by the originating UMTS

network, as well as the preferred codec. The originating side GMSC selects a codec from the transmitted list of codecs, and signals its selection back to the MSC in the Message (Selected Codec 1). Subsequently, a call connection is established at the bearer level (e.g. AAL2 or IP) with sufficient bandwidth to support the selected codec.

To enable end-to-end codec negotiation it is proposed to add the GCN to the ISUP protocol. This end-to-end codec negotiation will maximize the possibility of the end points utilizing the same codec type. If the end points use the same voice encoding algorithm, TFO is able to pass the compressed voice through the PCM network without degrading the voice quality due to unnecessary transcoding. This also enables transmission savings by minimizing the bearer requirements within the originating and terminating networks that support compressed voice (e.g. AAL2 or IP bearer transport).

Figure 3 illustrates the use of GCN enhanced ISUP messages to bridge the signalling "gap" between the two UMTS networks. At the terminating UMTS network, an IAM is sent using TICC from the GMSC to the MSC. The MSC in this case accepts use of codec 1 and signals this back to the GMSC again using TICC. Subsequently, the bearer level connection is established at the terminating UMTS network. The PSTN relays the codec acceptance to the originating side GMSC. As the originally proposed codec has been accepted, there is no need to change the bearer level connection at the originating UMTS network. However, if there is a change in the codec, this must be sent from the originating side GMSC to the MSC, so that the bearer level connection can be modified, e.g. to increase the bandwidth of the connection. In Figure 3, ACM indicates an Address Complete Message and ANM indicates an Answer Message.

Figure 4 illustrates a second embodiment of the invention. This solution relies on the optional Codec Mismatch Resolution and Optimisation procedure in the TFO protocol to detect incompatible codecs. When TFO detects codec incompatibility, it can trigger a Codec modification procedure to modify the codec used by the terminals. TFO specifies the rules for resolving codec mismatch (i.e. which codec to select). The TFO protocol then triggers the TICC signalling to modify the codec used on the call, to enable compatible codecs in the two end terminals. A change from the codec initially

suggested by the originating UMTS network may require a modification to the bearer level connection established in either UMTS network.

The embodiments described above minimise unnecessary speech transcoding due to intermediate PCM networks as well as allowing for the optimal allocation of user plane equipment (e.g. transcoding units) and/or user plane resources (e.g. bandwidth) to support the service level of a particular call in public telecommunication networks.

Negotiation of the codec type will be required in existing networks and new networks that support only PCM encoded voice in order to minimize unnecessary transcoding in the speech path. The UMTS and GSM subscriber's speech quality will not be degraded unnecessarily when their calls traverse existing PCM core networks having TFO support. When the GCN mechanism is introduced into the TICC protocol, it is likely to be carried in a transparent method (APM User).

It will be appreciated by the person of skill in the art that various modifications may be made to the above described embodiment without departing from the scope of the present invention.

CLAIMS:

1. A method of setting up a call connection between first and second mobile telephone networks at least one of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the method comprising the steps of:

conducting a negotiation between said TFO device of one of the mobile networks and a peer TFO device of the other mobile network to determine a suitable speech codec; and

notifying said Radio Access Network of the determined speech codec by sending a Call Control (CC) message from the associated TFO device.

2. A method according to claim 1 wherein each of the mobile networks comprises a Tandem Free Operation (TFO) device located outside of the Radio Access network part(s) of the mobile network.

3. A method according to claim 1 or 2, wherein the two mobile telephone networks are coupled to one another via a PSTN.

4. A method according to any one of the preceding claims, wherein said negotiation is carried out using the Codec Mismatch Resolution and Optimisation Procedure in the TFO protocol, with TFO messages being sent using in-band signalling.

5. A method according to any one of the preceding claims, wherein said the or each TFO device located outside of a Radio Access network is located at a Gateway MSC (GMSC) which provides an interface between the mobile network and foreign networks.

6. A method according to any one of the preceding claims, wherein the network in which the Tandem Free Operation (TFO) device is located outside of the Radio Access Network part(s) is a Universal Mobile Telecommunication System (UMTS) network, and the Radio Access Network is a UMTS Terrestrial Radio Access Network.

7. Apparatus for setting up a call connection between first and second mobile telephone networks at least one of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the apparatus comprising:

means for conducting a negotiation between said TFO device of one of the mobile networks and a peer TFO device of the other mobile network to determine a suitable speech codec; and

means for notifying said Radio Access Network of the determined speech codec by sending a Call Control (CC) message from the associated TFO device.

8. A method of setting up a call connection between first and second mobile telephone networks each of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which the Call Control protocol is independent of the bearer transport mechanism, the mobile networks being coupled via a Public Switched Telephone Network (PSTN), the method comprising the step of:

conducting a negotiation between two mobile terminals subscribing to the first and second mobile networks respectively to determine a suitable speech codec, wherein the negotiation is conducted using Call Control protocol signalling messages exchanged between the Radio Access Network parts and the respective TFO devices, and ISUP messages sent between the TFO devices.

9. A method according to claim 8, wherein a list of codecs which are available to an originating mobile network are sent from the TFO device of that network to the peer TFO device using an ISUP Initial Address Message (IAM) and the selected codec is returned to the originating network in a subsequent ISUP message.

10. Apparatus for setting up a call connection between first and second mobile telephone networks each of which comprises a Tandem Free Operation (TFO) device located outside of the Radio Access Network part(s) of the mobile network and in which

the Call Control protocol is independent of the bearer transport mechanism, the mobile networks being coupled via a Public Switched Telephone Network (PSTN), the apparatus comprising:

means for conducting a negotiation between two mobile terminals subscribing to the first and second mobile networks respectively to determine a suitable speech codec, wherein the negotiation is conducted using Call Control protocol signalling messages exchanged between the Radio Access Network parts and the respective TFO devices, and ISUP message sent between the TFO devices.

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13

Application No: GB 9915366.0 **Examiner:** Glyn Hughes
Claims searched: 1 to 10 **Date of search:** 11 January 2000

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.R): H4L (LDSC, LDLX)

Int Cl (Ed.7): H04Q 3/00, 7/24

Other: Online: WPI, JAPIO, EPODOC

Documents considered to be relevant:

Category	Identity of document and relevant passage	Relevant to claims
A	GB 2319144 A (NOKIA) see whole document	-
A	WO 96/23297 A1 (QUALCOMM) see whole document	-

X Document indicating lack of novelty or inventive step
Y Document indicating lack of inventive step if combined with one or more other documents of same category.
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E Patent document published on or after, but with priority date earlier than, the filing date of this application.